# WEST Search History

DATE: Friday, September 12, 2003

| Set Name side by side | Query | Hit Count | Set Name result set |
|---|---|---|---|
| | *DB=USPT,PGPB,JPAB,EPAB,DWPI,TDBD; PLUR=YES; OP=ADJ* | | |
| L9 | L8 and @AD<20000811 | 22 | L9 |
| L8 | (((contact or call) near8 center)) same (monitor or monitoring or monitored) same (allocate or allocation or allocated)same (agent or channel or resource) | 28 | L8 |
| L7 | l5 and (compare or comparison or compared) | 17 | L7 |
| L6 | (((contact or call) near8 center) or ACD) same (monitor or monitoring or monitored) same (allocate or allocation or allocated) | 45 | L6 |
| L5 | L4 and (parameter or attributr or threshold) | 23 | L5 |
| L4 | L3 and @AD<20000811 | 34 | L4 |
| L3 | (((contact or call) near8 center) or ACD) same (monitor or monitoring or monitored) same (allocate or allocation or allocated) | 45 | L3 |
| L2 | (((contact or call) near8 center) or ACD) same (monitor or monitoring or monitored) same (allocate or allocation or allocated) same (operation near8 (parameter or attribute)) | 0 | L2 |
| | *DB=USPT; PLUR=YES; OP=ADJ* | | |
| L1 | (((contact or call) near8 center) or ACD) same (monitor or monitoring or monitored) same (allocate or allocation or allocated) same (operation near8 (parameter or attribute)) | 0 | L1 |

END OF SEARCH HISTORY

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L7: Entry 2 of 17 File: USPT Dec 31, 2002

DOCUMENT-IDENTIFIER: US 6501736 B1
TITLE: System for increasing the call capacity of a wireless communication system

Abstract Text (1):
The system for increasing the call capacity of a wireless communications system degrades voice quality of the communications connections within a predetermined limit in order to increase the efficiency of the available RF spectrum as measured by the call carrying capacity of this allocated RF spectrum. This is accomplished by adjusting the transmission rate of the speech coder at the mobile subscriber unit and/or the speech coder that may be located at the mobile switching center so that the call carrying capacity of the wireless communications system is therefore increased. Additionally, the situation of call blocking is monitored to determine if frame error rate targets should be adjusted to further increase the call capacity. At times when the wireless system is not experiencing peak usage, the voice quality is restored to normal levels. The term "mobile subscriber unit" does not imply that the mobile unit is restricted to a moving vehicle. The present state of the art includes the use of mobile subscriber units in fixed wireless applications as an alternative to traditional wire telephony services.

Application Filing Date (1):
19990319

Brief Summary Text (8):
The above problem is solved and a technical advance achieved in the field of wireless communications systems by the present system for increasing the call capacity of CDMA channels. In this system, if call blocking is detected in the wireless communications system it may be acceptable to degrade voice quality of the communications connections within a predetermined limit in order to increase the efficiency of the available RF spectrum as measured by the call carrying capacity of this allocated RF spectrum. This is accomplished by adjusting the transmission rate of the speech coder at the mobile subscriber unit and/or the speech coder that may be located at the mobile switching center so that the call carrying capacity of the wireless communications system is therefore increased. Additionally, the situation of call blocking is monitored to determine if frame error rate targets should be adjusted to further increase the call capacity. At times when the wireless system is not experiencing peak usage, the voice quality is restored to normal levels.

Detailed Description Text (22):
When mobile subscriber unit 201 wishes to initiate a mobile-originated call, mobile subscriber unit 201 sends an origination message 205 to the appropriate base station 202. In response, base station 202 replies with a channel assignment message 206, which assigns mobile subscriber unit 201 to a traffic channel. All messaging between mobile subscriber unit 201 and base station 202 occurs on an associated radio channel. Subsequently in the set up of the call, base station 202 sends a service connect message 207 to the mobile subscriber unit 201 in order to configure the traffic channel. Mobile subscriber unit 201 returns a service connect complete message 208 to base station 202 to acknowledge the completion of configuring the traffic channel. Base station 202 determines that the total power (over the plurality of mobile subscriber units being served at the time of setting up the given call) exceeds a threshold on the reverse link. This determination is a function of the speech coder rate reduction subprocess, which is described later in this disclosure. Consequently, base station 202 sends a service option control order message 209 to the mobile subscriber unit 201. (If the heretofore mentioned threshold is not exceeded, message 209 does not need to be sent.) Message 209 causes the mobile subscriber unit to adjust the transmission rate of its speech coder, according to the ORDQ field 210. At this point of time and thereafter, the call is considered as an established call; otherwise the call is not established. At a later point of time in the call, base station 202 may determine that

the level of the speech coder reduction needs to be modified, resulting from a different measured total power on the reverse link. (The level corresponds to the levels in the heretofore mentioned tables relating the reduction level to the modified transmission rate.) This determination is again a function of the speech coder rate reduction subprocess. In such a case, base station 202 sends message 211 to the mobile subscriber unit 201.

Detailed Description Text (23):
FIG. 3 is analogous to FIG. 2 except that the affected speech coder is located at the mobile switching center 303. Messages 304, 305, 306, 307, and 308 have the same function as messages 204, 205, 206, 207, and 208, respectively. However, previous to base station 302 sending a speech coder reduction order message 309 to the mobile switching center 303, base station 302 determines that the total power (over the plurality of mobile subscriber units being served at the time of setting up a call) exceeds a threshold on the forward link. This determination is a function of the speech coder reduction process. Message 309 contains the ORDQ1 field 310 that is substantially equivalent in scope as the ORDQ field 210 that is contained in message 209. However, the corresponding values may be different for the total power on the forward link and the total power on the reverse link. Message 309 is transported from base station 302 to mobile switching center 303 using terrestrial facilities such as a T1 facility. At a later time in the call, when the call is established, base station 302 may determine that the level of speech coder reduction needs to be modified, resulting from a different measured total power on the forward link. In such a case, base station 302 sends message 311 to the mobile switching center 303.

Detailed Description Text (28):
FIG. 4 illustrates the flow diagram for the call capacity enhancement process. Step 401 initiates the process, which is a consequence of messages 204 and 304 (FIGS. 2 and 3, respectively) being received at the selected base stations. Step 402 determines if any parameters need to be adjusted. These parameters is discussed later and are determined by the speech coder rate reduction subprocess 405 and the adjust FER subprocess 407. Initially, these parameters are not adjusted since additional steps (as is discussed hereafter) must be executed. With further iterations of this process, the appropriate parameters are adjusted. In step 403, the appropriate parameters are updated. Parameters associated with the mobile subscriber unit are contained in a service option control order message (FIG. 2); parameters associated with the mobile switching center are contained in a speech coder reduction order message (FIG. 3); parameters associated with the base station are directed within the base station. In step 404, the base station determines if calls are being blocked. Calls may be blocked for a number of call scenarios, including mobile-originated call attempts, mobile-terminated call attempts, and handoff attempts. For CDMA, call blocking is a condition in which additional calls will degrade current calls more than a predetermined level. In other words, call blocking is not "hard" but is "soft". In the preferred embodiment, this is determined by comparing the total power on the reverse link to an associated maximum allowable level and the total power on the forward link to an associated maximum allowable level, which may be different from the level associated with the reverse link. If either of the total power levels exceed corresponding limits, the call capacity enhancement process determines that calls are being blocked. If so, the speech coder rate reduction subprocess 405 is executed. Subprocess 405 is discussed in greater detail. The result of subprocess 405 is that values of parameters ORDQ and ORDQ1 may be modified. If so, the modified parameters is stored and is adjusted when step 403 is executed in the next iteration of the call capacity enhancement process. If calls are blocked, as determined by step 406, the adjust FER (frame error rate) subprocess 407 is executed. Step 406 is substantially equivalent to step 404. Subprocess 407 utilizes symbol error based power control as described in U.S. Pat. No. 5,727,033, issued to Weaver et al., the full text of which is incorporated herein by reference as if reproduced in full. The result of executing subprocess 407 is to determine if the E.sub.b /N.sub.0 target needs to be adjusted. If so, the modified parameter is stored and is adjusted when step 403 is executed in the next iteration of the call capacity enhancement process.

Detailed Description Text (31):
The speech coder rate reduction subprocess is illustrated as a flow diagram in FIG. 5. Step 501 initiates the subprocess at the base station. Step 502 determines if the subprocess is activated based upon the time and date. Activation can be initiated by message 204 in FIG. 2. If the subprocess is activated, step 504 determines if the total power on the reverse link exceeds a threshold T4r. Threshold values, as discussed in the context of this subprocess as disclosed as the preferred embodiment, are entered as data entries. It is assumed that the following relationships exist among these

thresholds, namely T4r>T3r>T2r>T1r and T4f>T3f>T2f>T1f. If step 504 determines that the total power on the reverse link exceeds T4r, the mobile speech coders rate reduction parameter is set to 4 in step 512, and step 513 is executed in which the total power on the forward link is compared with threshold T4f. In step 504, if the total power on the reverse link is not greater than T4r, the total power on the reverse link is compared with threshold T3r. In step 505, if the total power on the reverse link is greater than T3r, the mobile speech coder rate reduction parameter is set to 3 in step 506 and step 513 is next executed. Otherwise, step 507 is executed in which the total power on the reverse link is compared to T2r. If greater, the mobile speech coder rate reduction parameter is set to 2 in step 508 and step 513 is executed. If not greater, the total power on the reverse link is compared to threshold T1r in step 509. If greater, the mobile speech coder rate reduction parameter is set to 1 in step 510 and step 513 in executed. If not greater, the mobile speech coder rate reduction parameter is set to 0 in step 511. In step 513, if the total power on the forward link is greater than T4f, then the MSC speech coder rate reduction parameter is set to 4 in step 514 and the subprocess is terminated in step 515. If not then the total power on the forward link is compared to threshold T3f in step 516. If greater, MSC speech coder rate reduction parameter is set to 3 in step 517. The subprocess consequently terminated in step 518. If not greater, the total power on the forward link is compared to T2f in step 519. If greater, then MSC speech coder rate reduction parameter is set to 2 in step 520 and the subprocess is terminated in step 521. If not, the total power on the forward link is compared to threshold T1f in step 522. If greater, MSC speech coder rate reduction parameter is set to 1 in step 523 and the subprocess is terminated in step 524. Otherwise, MSC speech coder rate reduction parameter is set to 0 in step 525 and the subprocess is terminated in step 526.

Detailed Description Text (33):
Thus, a wireless communications system compares the total power on both the forward link and the reverse link with a set of thresholds as determined by the service provider either by manual or automated means. Consequently, the transmission rate of the speech coder at the mobile subscriber unit and of the speech coder at the mobile switching center as well as the target values for the frame error rate are adjusted accordingly. This adjustment enables a wireless communications system to increase its call capacity without additional spectrum.

CLAIMS:

1. In a wireless communications system that provides wireless communication services to a plurality of wireless subscriber units extant in said wireless communication system, by utilizing a plurality of variable rate speech coders that have at least one adjustable operating parameter associated with a forward radio channel and having at least one adjustable operating parameter associated with a reverse radio channel, apparatus for minimizing a total power on at least one of said forward radio channel and said reverse radio channel, comprising: means for determining an occurrence of call blockage, comprising: means for measuring a total power summed over all of said wireless subscriber units being served on said forward radio channel, means for comparing said total power on said forward radio channel to at least one threshold associated with said forward radio channel in order to determine if said total power on said forward radio channel is greater than at least one of said at least one threshold associated with said forward radio channel, means for measuring a total power summed over all wireless subscriber units being served on said reverse radio channel, means for comparing said total power on said reverse radio channel to at least one threshold associated with said reverse radio channel in order to determine if said total power on said reverse radio channel is greater than at least one of said at least one threshold associated with said reverse radio channel; and means, responsive to said occurrence of call blockage, for adjusting at least one of said at least one adjustable operating parameter associated with at least one of said forward channel and said reverse channel, whereby a resulting call carrying capacity of said wireless communications system is increased.

2. The wireless communications system in claim 1 further comprises: means for identifying wireless subscriber units that are assigned a level of quality of service that Is less than a predetermined threshold; and wherein said means for adjusting is operable to adjust said at least one of said at least one adjustable operating parameter only for said identified wireless subscriber units.

5. The wireless communications system in claim 1, wherein said means for adjusting said at least one of said at least one adjustable operating parameter associated with said forward radio channel comprises: means for reducing a speech coder transmission rate

according to said threshold associated with said forward radio channel that is exceeded by said total power summed over all of said wireless subscriber units being served on said forward radio channel.

6. The wireless communications system in claim 5, herein said means for adjusting at least one of said at least one adjustable operating parameter associated with said reverse radio channel comprises: means for reducing a speech coder transmission rate according to said threshold associated with said reverse radio channel that is exceeded by said total power summed over all of said wireless subscriber units being served on said reverse radio channel.

15. In a wireless communications system that provides wireless communication services to a plurality of wireless subscriber units extant in said wireless communication system, by utilizing a plurality of variable rate speech coders that have at least one adjustable operating parameter associated with a forward radio channel and having at least one adjustable operating parameter associated with a reverse radio channel, a method for minimizing a total power on at least one of said forward radio channel and said reverse radio channel, comprising the steps of: determining an occurrence of call blockage comprising: measuring a total power summed over all said wireless subscriber units being served on said forward radio channel, comparing said total power on said forward radio channel to at least one threshold associated with said forward radio channel in order to determine if said total power on said forward radio channel is greater than at least one of said at least one threshold associated with said forward radio channel, measuring a total power summed over all said wireless subscriber units being served on said reverse radio channel, comparing said total power on said reverse radio channel to at least one threshold associated with said reverse radio channel in order to determine if said total power on said reverse radio channel is greater than at least one of said at least one threshold associated with said reverse radio channel; and adjusting, in response to said occurrence of call blockage, at least one of said at least one adjustable operating parameter associated with at least one of said forward channel and said reverse channel, whereby a resulting call capacity of said wireless communications system is increased.

16. The method for minimizing said total power of claim 15 further comprising the steps of: identifying wireless subscriber units that are assigned a level of quality of service that is less than a predetermined threshold; and adjusting at least one of said at least one adjustable operating parameter only for said identified mobile subscriber units.

19. The method for minimizing said total power of claim 15, wherein said step of adjusting at least one of said at least one adjustable operating parameter associated with said forward radio channel further comprises: reducing a speech coder transmission rate according to said threshold associated with said forward radio channel that is exceeded by said total power summed over all said wireless subscriber units being served on said forward radio channel.

20. The method for minimizing said total power of claim 15, wherein said step of adjusting at least one of said at least one adjustable operating parameter associated with said reverse radio channel further comprises: reducing a speech coder transmission rate according to said threshold associated with said reverse radio channel that is exceeded by said total power summed over all said wireless subscriber units being served on said reverse radio channel.

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L7: Entry 3 of 17 File: USPT Nov 5, 2002

DOCUMENT-IDENTIFIER: US 6477667 B1
TITLE: Method and system for remote device monitoring

Abstract Text (1):
A user contracts for service with an operations center (12) in order to provide monitoring services for a plurality of devices (30). After contracting for service, the operations center provides an agent (81) for download by a user to one or more of the user's devices (14, 16, 18, 20, 22) for which the user has contracted for service. The agent is installed on the devices associated with the user's sites and a listener (362) at the operations center listens for alerts generated as a result of the agent monitoring health-indicative operating parameters on the device. After an alert is received by the operations center, a contact (32) is notified of the alert so that the problem causing the generation of the alert may be corrected.

Application Filing Date (1):
19991007

Brief Summary Text (6):
According to the present invention, a method and apparatus are provided to address this need, and involve a monitoring system which includes a communications network and an agent unidirectionally coupled to the communications network and residing at a remote site. The agent is operable to monitor a set of operating parameters. The agent is further operable to generate an alert in response to an operating parameter exceeding a predetermined threshold and to transmit the alert across the communications network. The monitoring system further includes a listener coupled to the communications network and operable to receive the alert, and a responder operable to act in response to the alert.

Detailed Description Text (4):
Each site 14, 16, 18, 20 and 22 may include one or more devices 30. Hereinafter sites will be referred to generally as "site or sites 14" with the other reference numbers (16, 18, 20 and 22) being used to refer to particular sites. Sites 14 may represent physical and logical entities that have contracted with operations center 12 for monitoring services. Site 14 may be a company, a department within a company, a building, a geographic area, a logical entity occupying multiple geographic locations, or other suitable logical or physical entities capable of being monitored-over Internet 34 from operations center 12. The monitoring services provided by operations center 12 may include the monitoring of various operating parameters or predetermined status indicators (not shown) which indicate the present or predicted future health of devices 30 being monitored. The process for contracting for service will be described in more detail in association with FIG. 2.

Detailed Description Text (5):
Device 30 may be any of a plurality of electronic devices having simple or advanced data processing capabilities and health-indicative operating parameters that may be monitored by and communicated to a remote location, such as operations center 12. Each device 30 is associated with at least one site 14. For example, device 30 may be a server, a workstation, a personal computer, a laptop, a soft drink dispensing machine, a network postage machine, a printer, a personal digital assistant, a heating/ventilation/air conditioning (HVAC) system or another suitable device. Health-indicative operating parameters are status indicators which may be used to determine the current or predicted future operational status or health of device 30. The operating parameters, for example, may indicate that device 30 could cease operating in the near future, that device 30 is operating slower or less optimally than expected, that device 30 is more heavily loaded with processing requests than it should be, that the persistent storage associated with device 30 may be failing, and that device 30 is running out of supplies and inventory, such as cans of soft drink or a

printer running out of ink and paper. Other status indicators associated with device 30 that may be utilized in the repair, debugging or monitoring of device 30 may also be used.

Detailed Description Text (6):
The operating parameters may vary based on the particular device 30 being monitored. For example, if device 30 being monitored is a network postage meter then the operating parameters may include the remaining postage available on the meter and whether the remaining postage has fallen below a particular level, whether the amount of ink is low, and whether the system is operational.

Detailed Description Text (7):
The health indicative operating parameters may vary based on the operating system and hardware used by device 30. Generally, the health indicative operating parameters may monitor the available disk space for a particular user, the number of failed log-in attempts for one or more users, the number of license connections currently available on the server and the network traffic load on the server.

Detailed Description Text (8):
For example, if device 30 utilizes the Windows 95/98 operating system, the health indicative operating parameters may include the available dynamic memory and whether it has fallen below a particular threshold, the processor utilization percentage and whether the utilization exceeds or drops below a particular threshold for a specific amount of time, system errors, general protection faults, system reboots, the relay of an event from a proprietary protocol, such as the Compaq Insight Manager, and the number of bad blocks on a hard drive. The health indicative operating parameters may further include the available memory, CPU utilization, available disk space, available system resources, available graphics device interface (GDI) resources, available user resources, whether the hard drives are on-line, and information regarding system start-up. The available memory may represent the percentage of total memory that is not being used, the CPU utilization may represent the percentage of time that the CPU is not idle, the available disk space may include the percentage of the total disk space that is not being used on each logical and physical hard drive on device 30, the available system resources may include the percentage of the total system resources that are not being used, the GDI resources may include the percentage of the total GDI resources that are not currently in use, and user resources may include the percentage of the total user resources that are not being used.

Detailed Description Text (9):
If device 30 is using the Windows NT operating system, then the health indicative operating parameters may include the available memory, the CPU utilization for each CPU, the available hard disk space, whether the hard drive is online, information regarding system start-up, event log alerts, application log alerts, Internet Information Service status, Structured Query Language service status and security log alerts. The logs may be monitored for particular alerts or information and generate alerts based on that information. The available memory may include the percentage of total memory, physical memory and virtual memory, either individually or as a group, that is not being used. The CPU utilization for each CPU may include the percentage of time each CPU is not in an idle state, and the available hard disk space may include the percentage of unused space on each logical and physical hard drive.

Detailed Description Text (10):
If device 30 is using the Novell Netware operating system, the health indicative operating parameters may include the available cache buffers, the CPU utilization, the available disk space, volume status, system start-up information, the number of purgable blocks on a volume, the forged pack count and a count,of invalid sequence numbers. The available cache buffers may include the percentage of the total cache buffers that are not being used, the CPU utilization may represent the percentage of time that the CPU is not idle, the available disk space may represent the percentage of unused space on each volume managed by the Novell Netware operating system, and the volume information may include whether a particular volume is on-line and operating.

Detailed Description Text (14):
FIG. 2 is a flow diagram of a process for signing up with or contracting for service with remote device monitoring system 10. The sign-up process is initiated and performed by a user 45 in order to contract for monitoring service from remote device monitoring system 10. The user 45 is a human user of the present invention. The sign-up process is used to determine and allocate the number of licenses the user 45 will require, as well as acquire information from the user 45. In the disclosed embodiment, two types of

licenses are available, a server license and a workstation license. The server license is required for each server the user 45 wishes to have monitored by system 10 and the workstation license is required for each workstation, PC or other non-server device that the user 45 wishes to be monitored. It should be noted that both servers and non-server devices may be devices 30. In the disclosed embodiment, the sign-up process is initiated by retrieving a web page 50 associated with remote device monitoring system 10 using a web browser (not shown). Web page 50 may comprise a plurality of web pages and may be stored on a web server (not shown) at operations center 12 or another suitable location. The sign-up process may be initiated in other ways, such as by a telephone call or electronic mail to operations center 12 or a service center (not shown), or by another suitable method by which the necessary information for server and workstation licenses may be obtained. Regardless of the contact method used, the sign-up process and the necessary information remain substantially similar.

Detailed Description Text (31):
Next, at block 80, an agent 81 is deployed to the device or devices 30 that the user 45 has registered for monitoring in block 72. In the disclosed embodiment, agent 81 is a file which is downloadable from a server via the file transfer protocol (FTP) or HTTP and is a C++ based operating system extension specific to a particular operating system, but may be an applet or application written in any suitable platform-independent programming language such as C, Java and Perl. Agent 81 may be installed on device 30 automatically or manually by the user 45 and performs the actual monitoring of device 30. In particular, agent 81 tracks the various operating parameters which are used to determine the current health of device 30 and generates alerts when device 30 may be experiencing problems. The operation of the agent is described in more detail below in association with FIG. 6.

Detailed Description Text (53):
FIG. 6 is a flow diagram showing details of the generation and processing of an alert for device 30. Agent 81 associated with device 30 operates to monitor various health-indicative operating parameters associated with each device 30. The operation of agent 81 is described in more detail in FIG. 6A. When one of the health-indicative parameters exceeds a predetermined threshold or value indicative of poor health, or indicating a high likelihood of poor health or failure, agent 81 generates an alert in block 360. The alert includes the device identifier and MAC address of device 30 which is generating the alert. The alert also includes version information associated with agent 81 so that operations center 12 may notify the user that a new version of agent 81 is available. The alert is then unidirectionally transmitted over Internet 34 by agent 81 to a listening process 362. In particular, agent 81 is operable only to transmit information outbound from the device 30 and provides no support for receiving inbound information or connections. By allowing agent 81 to only transmit outbound information, greater security is maintained for device 30 and site 14 as no additional entry points are provided for exploitation by hackers and intruders.

Detailed Description Text (61):
At block 404, agent 81 performs set up and initialization procedures such as allocating needed memory and initializing variables. At block 406, agent 81 loads current values for all indicators, such as the health indicative operating parameters previously described, into MIB variables for use with SNMP.

Detailed Description Text (62):
At block 408, agent 81 performs blocks 410, 412, 414, and 416 for each health indicative operating parameter. At block 410, the current operating parameter is examined to determine its current value on device 30 and is compared to a threshold value. The SNMP alerts may be transmitted to operations center 12 using the universal datagram protocol (UDP). At block 412, an alert is generated if the current value of the current operating parameter is; outside the threshold values or other values within which the current operating parameter is expected to operate. If the current operating parameter is outside of its allowed operational range, then the YES path of decisional step 412 is followed and an SNMP alert is created for the current operating parameter in block 414. The SNMP alert may include an SNMP variable binding list containing the appropriate MIB variables associated with the operating parameter, the MAC address of device 30 and the device identifier associated with device 30. If the current operating parameter has not exceeded its allowed operational range, then the NO branch of decisional step 412 will be followed to block 416. At block 416, the next operational parameter is set as the current operational parameter and the method returns to block 410.

Detailed Description Text (63):

Once each operating parameter has been examined in block 408, the method proceeds to decisional step 418. At decisional step 418, a check is made to see if any SNMP traps have been created in block 414. If any traps have been generated, then the YES branch of decisional step 418 is followed to block 420 where the generated traps are treated as alerts and sent to operations center 12. If no traps have been generated in block 414, then the NO branch of decisional step 418 is followed to block 422. At block 422, agent 81 may sleep for a predetermined period of time, such as five seconds, and then continue to step 406 to again check the current values of each operating parameter on device 30. The method proceeds until agent 81 is terminated, such as at system shutdown.

Detailed Description Text (74):
The present invention provides a number of technical advantages. One such technical advantage is the capability for unidirectional monitoring of the health of a device. The unidirectional monitoring allows for the use of unsecured communications links, such as the Internet, between the monitored device and an operations center. Unidirectional monitoring avoids the requirement for installing and maintaining expensive secure communications lines. An agent deployed to devices at a site is operable only to transmit information from the device to the operations center which prevents the use of the agent by intruders and hackers to gain access to the site. A further technical advantage is the capability of the present invention to provide remote and scalable monitoring of the health of devices inexpensively. Small and medium sized businesses may now take advantage of the service with minimal cost compared to traditional comprehensive system management.

CLAIMS:

1. A monitoring system comprising: an agent unidirectionally coupled to a communications network and residing at a remote site, the agent operable to monitor a set of operating parameters and to generate an alert in response to an operating parameter exceeding a predetermined threshold and transmit the alert across the communications network; wherein the agent is further operable to generate a registration trap when deployed at a device associated with the site; a listener coupled to the communications network and operable to receive the alert; and a responder operable to act in response to the alert.

4. The monitoring system according to claim 1, wherein the operating parameters comprise: an available hard disk storage space indicator; and a hard disk media error indicator.

5. The monitoring system according to claim 1, wherein the operating parameters comprise: an available memory indicator; and an available system resources indicator.

6. The monitoring system according to claim 1, wherein the operating parameters comprise: a processor utilization level indicator; and a system error indicator.

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L7: Entry 4 of 17 File: USPT Mar 26, 2002

DOCUMENT-IDENTIFIER: US 6363145 B1
TITLE: Apparatus and method for automated voice analysis in ACD silent call monitoring

Abstract Text (1):
A method and system for automated silent call monitoring in an automatic call distributor (ACD) environment includes configuring a set of call performance profiles which include voice data patterns which are descriptive of voice data transmissions during an ACD call and which are associated with substandard agent performance. Each voice data pattern has a corresponding threshold which represents the maximum number of detections tolerated in an ACD call prior to execution of a notification routine. A digital signal processor (DSP) monitors a first call between an agent terminal and a customer terminal for the voice data patterns and stores detection data in memory upon detecting of one of the voice data patterns. A central processor unit (CPU) compares the number of detections of the voice patterns within predetermined time intervals to the threshold numbers of detections represented in the thresholds to determine if any threshold has been exceeded. If no thresholds have been exceeded, the DSP continues to monitor for the voice data patterns. If a threshold has been exceeded, the CPU executes the notification routine wherein a supervisor terminal and the agent terminal are notified of the exceeded threshold. A supervisor notification can include an option to establish a direct monitoring session for the first call and an option to transfer the first call to the supervisor terminal. An agent notification preferably includes call performance analysis which provides suggestions for improving call performance which are responsive to the detected voice data patterns.

Application Filing Date (1):
19980817

Brief Summary Text (5):
If the ACD supervisor becomes familiar with the ACD agents whom the supervisor oversees, the supervisor is able to allocate his time effectively by spending a majority of time on those ACD agents whose skills require more development and spending less time supervising those ACD agents whose skills are already refined. However, if an ACD supervisor is not familiar with the ACD agents he supervises, either because of high agent turnover or because he is new to the job, the supervisor does not have the proper information to focus attention on those ACD agents who most urgently require it. Consequently, the supervisor wastes time monitoring agents who do not require supervision, while agents who urgently require supervision do not receive the attention they require.

Brief Summary Text (10):
A method and system for silent ACD call monitoring utilizing automated voice analysis includes identifying multiple voice data patterns associated with substandard agent performance, monitoring a first call between an ACD agent terminal and a customer terminal to detect the voice patterns, determining whether the number of occurrences of any one of the voice data patterns exceeds an associated predetermined threshold, and notifying the agent terminal and/or a supervisor terminal upon detecting that a threshold number of voice data pattern occurrences has been exceeded.

Brief Summary Text (12):
A processor, such as a digital signal processor (DSP), is utilized to detect the voice data patterns associated with substandard performance. The voice data patterns include a length of silence in conversation between the customer and the agent in excess of a predetermined length which indicates either poor information delivery by the agent or lack of interest on the part of the customer. A conversation volume above a maximum volume level tends to indicate a high frustration level in either the customer or the agent. Changes in voice frequency in excess of a predetermined range during the conversation tend to indicate an emotional exchange. The average length of continuous

conversation by the agent also provides information about agent call performance. If the average length of continuous agent conversation is above a maximum threshold, this tends to demonstrate that the agent is talking without paying sufficient attention to the customer. If the average length of continuous agent conversation is below a minimum threshold, this tends to show that the agent is not being responsive to the customer. Interruptions in conversation tend to indicate poor agent performance as well. An interruption of the agent by the customer demonstrates that the customer is frustrated and dissatisfied with the agent's responses, while interruptions by the agent demonstrate that the agent is being impatient and not listening to the customer.

Brief Summary Text (13):
Memory stores voice data pattern thresholds associated with the voice data patterns, such that each voice data pattern has a corresponding threshold. A threshold identifies a maximum number of occurrences of a corresponding voice data pattern which is tolerated during the first call between the agent and the customer. The thresholds can be configured to permit a predetermined number of occurrences for any given time interval during the first call and the number of occurrences permitted by thresholds associated with different voice patterns can differ. For example, the threshold number of agent interruptions per five-minute interval might be greater than the threshold number of occurrences of agent speech having volume above a predetermined level.

Brief Summary Text (14):
The number of occurrences of the voice data patterns are recorded by a counting device, the function of which can be performed by the DSP or a central processing unit (CPU). The number of detected occurrences of the voice data patterns is compared to the corresponding thresholds to determine whether any of the voice data patterns have been detected in excess of the corresponding threshold number of times. The comparison function can also be performed by either the DSP or the CPU. Upon discovering a voice data pattern which has been detected in excess of a threshold number of times, the CPU or DSP executes a notification routine to provide notification to the supervisor terminal and the agent terminal.

Brief Summary Text (15):
The monitoring of the first call for voice data patterns can occur at a different location from where the threshold comparisons take place. For example, the monitoring might take place at the agent terminal which transmits each detection to the supervisor terminal. The supervisor terminal calculates the total number of occurrences and determines whether any thresholds have been exceeded. When one of the thresholds is exceeded, the supervisor CPU causes a notification to be displayed on the supervisor terminal. The CPU can be configured to cause the notification device to include an option to establish a direct silent monitoring session with the agent terminal under specific circumstances. For example, if the agent's performance is poor, the notification can include an option to the supervisor to establish a direct monitoring session. The notification can also include an option to transfer the call from the agent terminal to the supervisor terminal if the agent's performance is determined to be particularly poor.

Brief Summary Text (16):
The notification routine can also provide notification to the agent terminal. In a preferred embodiment, the agent notification takes the form of performance analysis data which pertains to a detected voice pattern. For example, if the agent has interrupted the customer in excess of a threshold number of times, a message is displayed on the agent terminal which says "Avoid interrupting the customer." Alternatively the message can simply inform the agent of the number of times that the agent has interrupted the customer.

Detailed Description Text (4):
Referring to FIG. 2, the gateway 16 of the automated silent call monitoring system includes a LAN card 24 to enable communication between the first and second agent terminals 10 and 12 and the supervisor terminal 14. A basic rate interface (BRI) card 32 enables communication between the ISDN phone 20 and the first agent terminal 10, the second agent terminal 12 and the supervisor terminal 14. A digital signal processor (DSP) 22 analyzes agent voice data transmitted from either the first 10 or second agent terminal 12 and analyzes customer voice data transmitted from the ISDN phone 20 to detect the voice data patterns associated with poor customer service. The calculation processor 28, the comparison processor 30, and the DSP 22 have been illustrated in FIG. 2 as separate devices for purposes of clarity. However, the functions of all three can be incorporated into a single processor.

Detailed Description Text (6):
When the DSP 22 detects one of the voice data patterns, the DSP enters data representing the detection into memory 26. The calculation processor 28 accesses the detection data and updates the calculation of the number of detections that have occurred in an ACD call during a predetermined time interval. The predetermined time interval can be configured differently for each type of voice data pattern. For example, the calculation processor might calculate the total number of customer interruptions by the agent within a five-minute interval, whereas the total number of detections of occurrences of a voice exceeding the predetermined volume level might be calculated over a ten-minute interval. Logically, the more urgent the detection of a particular voice pattern is, the smaller the interval will be over which the number of detections is calculated, because a number of threshold detections for the more urgent voice pattern will be more quickly identified at the outset of the call. For example, detection of voice volumes over a certain high volume level indicates that communication in the call has severely deteriorated such that a single detection might require urgent attention. The time interval associated with such an event might be as short as 10 seconds.

Detailed Description Text (7):
After the calculation processor 28 updates the number of detections of a voice pattern, the comparison processor 30 accesses threshold data from memory 26 to determine whether a threshold number of detections has been exceeded. The parameters of the voice data pattern thresholds can be configured according to the requirements of the system user. For example, the tolerance for interruptions for a technical support agent will most likely be higher than for a catalog sales agent, because the technical support agent might be required to direct the conversation to a greater extent than the catalog sales agent. The comparison processor 30 compares the number of detections of the voice data pattern with the threshold number of detections indicated by the threshold data corresponding to that voice data pattern. If the comparison processor 30 determines the threshold has been exceeded, it causes a notification routine to be executed to provide notification to the supervisor terminal 14 and one of the agent terminals 10 and 12.

Detailed Description Text (8):
Although the automated silent call monitoring is shown in FIG. 2 as being performed in the gateway, it can be performed at the agent terminals 10 and 12, the gateway 16, the supervisor terminal 14, or the monitoring can be distributed amongst these devices. For example, the DSP 22 can be located in the agent terminal where it monitors a call to the ISDN phone 20 for the voice data patterns. If the DSP detects a voice data pattern, it can transmit data representative of the detection to the supervisor terminal 14, where it is entered into memory 26. The calculation processor 28 and the comparison processor 30, located in the supervisor terminal, process the detection data to determine if the corresponding voice data pattern threshold has been exceeded.

Detailed Description Text (11):
Separately monitoring agent voice data and customer voice data for voice patterns is important because a particular voice pattern can have different meanings depending on whether the voice pattern was detected in the agent voice data or the customer voice data. Detection of customer voice data above a predetermined threshold might be more readily tolerated than detection of agent voice data over the same predetermined threshold. For example, if the ACD is setup to provide customer technical support, the customers are often irate at the outset of the call as a result of the technical problem for which they are calling. The threshold number of occurrences of the customer speaking at a volume above the predetermined volume level might be relatively higher than the threshold number of occurrences of the agent speaking above the predetermined volume level.

Detailed Description Text (12):
When a voice data pattern has been detected in excess of the threshold number of times, the supervisor terminal 14 is notified of the event. The comparison processor 30 accesses supervisor notification data from memory 26 for transmission to the supervisor terminal. The notification can be in the form of a message displayed on a monitor of the supervisor terminal, reflecting detection of the voice data pattern in excess of the threshold number of occurrences. The notification can also include an option to the supervisor to establish a direct monitoring session, so that the supervisor can directly monitor the agent's performance for the remainder of the call. The comparison processor can be configured to provide the supervisor terminal with an option to transfer the first call from the first agent terminal 10 to the supervisor terminal 14, if the automated monitoring session indicates that the agent's call performance is particularly poor. For instance, if the agent is yelling at the customer and

interrupting the customer as well, the message might include an option to transfer the call from the first agent terminal 10 to the supervisor terminal 14.

Detailed Description Text (13):
The comparison processor 30 can also be configured to access agent notification data from memory 26 for presentation at the one of the agent terminals 10 and 12 in response to detecting occurrences of a voice data pattern in excess of the threshold number of times. The agent notification can simply indicate the call performance error associated with the voice data pattern, or it can include call performance analysis which provides suggestions for the agent to improve the agent's call performance. For example, if the agent has interrupted the customer in excess of the threshold number of times, the notification might be in the form of a message on the agent terminal's monitor that says "Avoid interrupting the customer."

Detailed Description Text (14):
Referring to FIGS. 2 and 3, a method for automated silent call monitoring includes the step 34 of configuring call performance profiles. Each call performance profile includes a voice data pattern which is descriptive of a characteristic of poor agent service during an ACD call with a customer, and at least one corresponding voice data pattern threshold which represents the maximum number of times the voice data pattern may be detected within a predetermined time interval before a notification procedure will be executed. Some call performance profiles might include two voice data pattern thresholds, including an agent threshold and a customer threshold. The voice data patterns include interruption in conversation by the agent and the customer, speech by the agent or the customer in excess of a predetermined volume level, fluctuations in the pitch of the voice of the customer or the agent in excess of a predetermined range, intervals of silence in the conversation in excess of a predetermined interval, and intervals of continuous agent or customer speech in excess of a maximum interval and below a minimum interval.

Detailed Description Text (15):
In step 36, agent notification messages are configured which will be presented at the agent terminal upon detection of one of the voice data patterns in excess of the threshold number of times. The agent notification messages preferably contain call performance analysis data providing suggestions to improve agent call performance.

Detailed Description Text (16):
In step 38, a first call is established between a first ACD agent terminal 10 and a customer phone 20. A DSP 22 monitors the first call in step 40 to detect any of the voice data patterns and in step 42, the DSP records detected voice data patterns into memory 26. In step 44, a comparison processor 30 determines whether the number of recorded voice data pattern detections for any of the voice data patterns exceeds a threshold. If no threshold is exceeded, the call continues to be monitored in step 40. The comparison processor 30 can also be configured to compare the number of voice data pattern detections between multiple calls to determine which call requires more urgent attention. For example, if a second call were established between the second agent terminal 12 and a second ISDN phone (not shown), the DSP 22 would simultaneously monitor both calls. If a threshold number of detections occurred in both the first and second calls, the comparison processor 30 can be configured to compare the number of detections in the first and second calls to determine which call requires more urgent attention. The specific types of voice data patterns that were detected in the first and second calls can also be considered in determining which call requires more urgent attention.

Detailed Description Text (17):
If a threshold number of detections is exceeded for any of the voice data patterns, in step 50 a notification routine is executed. The notification routine includes presenting a message at a supervisor terminal 14 which reflects detection of the threshold number of voice pattern occurrences in the first call. The message can also include an option to establish a direct monitoring session between the first agent terminal 10 and the supervisor terminal 14 and an option to transfer the call from the first agent terminal to the supervisor terminal. The notification routine further includes presenting a notification message at the agent terminal 10. The agent notification message includes call performance analysis data which includes suggestions for improving agent call performance.

CLAIMS:

1. A method for automated silent call monitoring utilizing automated analysis of speech

characteristics comprising the steps of: configuring call performance profiles representative of a plurality of voice patterns, with each said voice pattern being associated with a predetermined threshold number of occurrences; monitoring a first call between an agent at an agent terminal and a customer at a customer terminal for occurrences of said plurality of voice patterns during a conversation between said agent and said customer; maintaining a record of a number of occurrences for each of said voice patterns during said conversation; determining whether a number of occurrences of a particular one of said voice patterns exceeds said predetermined threshold number associated with said particular one; and initiating a notification routine upon detecting said number of occurrences exceeding said predetermined threshold number.

11. A system for performing silent call monitoring utilizing automated voice pattern detection to monitor a first call between an agent at an agent terminal and a customer at a customer terminal in an automatic call distributor (ACD) environment comprising: voice monitoring circuitry for detecting substandard performance of an agent user of said agent terminal during said first call, said voice monitoring circuitry comprising: a) means for detecting a plurality of voice data patterns associated with said substandard performance of said agent; b) memory including voice data pattern thresholds corresponding to said voice data patterns such that each voice data pattern has a corresponding threshold; c) means for comparing a total number of detected occurrences of a voice data pattern to a corresponding threshold; and notification means for providing notification upon detection of a number of said occurrences of one of said voice data patterns which is in excess of said corresponding threshold of said one.

13. The system of claim 11 wherein said comparing means is configured to compare said total number of detected occurrences within a predetermined time interval to said corresponding threshold.

15. The system of claim 14 wherein said memory further includes call performance analysis data associated with said voice data patterns such that each voice data pattern has associated call performance analysis data, said notification means being configured to present said call performance analysis data at said agent terminal upon detection of occurrences in excess of said threshold number of voice data pattern occurrences.

16. The system of claim 14 wherein said voice monitoring circuitry is configured to simultaneously detect substandard performance of a plurality agents, said comparing means being further configured to compare detected occurrences of said voice data patterns in said first call with detected occurrences of said voice data patterns in a second call to determine a relative urgency associated with said first and said second calls.

17. A method for silent monitoring of a first call between a customer and an ACD agent in an ACD system comprising the steps of: configuring voice pattern profiles to be descriptive of a plurality of voice patterns associated with substandard agent call performance wherein each voice pattern profile includes an associated threshold number of detections, said voice patterns including at least one of a length of silence in conversation between said agent and said customer in excess of a predetermined time interval, a conversation volume in excess of a predetermined volume level, voice frequency changes in excess of a predetermined fluctuation range, a continuous agent conversation interval above a maximum interval, and a length of continuous agent conversation below a minimum length; monitoring said first call for occurrences of said voice patterns; recording detections of said voice patterns in said first call; determining whether a number of detections of one of said voice patterns exceeds an associated threshold; and if said number of detections exceeds said associated threshold, executing a notification routine which provides notification of said detections to at least one of said agent terminal and an ACD supervisor terminal.

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L7: Entry 6 of 17 File: USPT Dec 11, 2001

DOCUMENT-IDENTIFIER: US 6330313 B1
TITLE: Communications network

Application Filing Date (1):
19980305

Detailed Description Text (5):
As illustrated in FIG. 1, a call answering centre 6 is connected to a DLE 5. In this example, the call answering centre has a single telephone number, e.g. 0800 400 496 and the capacity to handle 50 simultaneous calls. Although for ease of illustration only a single call answering centre is shown, in practice the SCP may be connected, via DSMU's and DLE's, to many such call answering centres. For example, it is envisaged that in the UK PSTN one SCP/ACS will support several hundreds of answering centres. In the ACS, an instance of a call control process for a given call answering centre is created only when the call answering centre becomes congested. To avoid the signalling overheads which would be incurred if every call were monitored, the ACS samples calls intermittently. For example, the ACS may select one call every 10 seconds. For the selected call, the ACS sets a detection point at the originating SSP which is triggered if a BUSY signal is returned from the DLE for a call to the answering centre number, 0800 400 496. In response to the trigger, an instance of the call control process is created for that number. As further described below, a counter which records the total number of calls in progress to the number is initiated. An estimate is made of the total capacity of the centre, and this value is stored. This estimate may be derived in an initial training phase, prior to the call control process taking control over whether calls are admitted. After the completion of the training phase, when a subsequent call is made to the answering centre, for example from a subscriber connected to DLE 4, a request for processing is passed from the service control function in the SCP to the answering centre server. The answering centre server compares the value for the capacity of the server with the value of the counter. If the total number of calls in progress (excluding the new call) is less than the value for the capacity of the answering centre, then the call is admitted and the counter is incremented to reflect the additional call in progress. This is signalled by the answering centre server to the service control function. The call is then progressed conventionally. At the same time, the answering centre sets detection points for the following termination events: busy, RTNR (ring tone, no reply), Abandon, Answer, Disconnect, Route Select Failure. When any of these termination events occur, this is signalled via the SCF/ACS interface, and the counter is decremented by one. Whenever a further call is received, these steps are repeated, and if the call is admitted the counter is again incremented. Further iterations of this process are carried out, until the number of calls in progress corresponds to the value for the capacity of the answering centre. Then when a further call is received, the condition that the number of calls in progress should be less than the capacity is not met. In this case, the server causes the SCP to send a ReleaseCall message to the originating exchange. This message may include a reason for the failure to complete the call, namely user busy. It should be noted that by contrast with the functioning of a conventional network, in which the BUSY signal would have been returned from the destination DLE, the call never progresses beyond the originating SSP and the SCP. The call does not make any contribution to the traffic in the destination exchange, and so infrastructure at the destination exchange does not have to be designed to support calls originating in these circumstances.

Detailed Description Text (34):
FIG. 8 illustrates the principal functional components of the ACS. A counter controller 81 is connected to a counter 82 and to a network signalling interface 83. A call controller 84 includes a comparator 86 which compares the value of the counter 82, and a call capacity value which is recorded in store 85. The call controller uses a control signal generator 87 to return a control signal via the network interface depending on

the results of the comparison. Although dedicated hardware may be used for some or all of these components, more usually the components are embodied by an appropriately programmed microprocessor and associated RAM (random access memory). An example of suitable software for implementing the ACS is contained in the Appendix below.

Detailed Description Text (35):
The OCS in this example is arranged to use a number of other resources, in addition to the ACS, to monitor and control traffic levels. In particular, it also uses Automatic Call Restriction (ACR) on the outbound traffic. As described in our above-cited international application, ACR limits the rate of calls admitted to the answering centre, to a value rather higher than the answering centre's answering rate capability. It does not need to know the answering centre's answering rate capability in order to do this: a teletraffic result states that for a call holding time of 10 s or greater, an excess of arrivals over mean answering capacity of 1.8 calls s.sup.- is sufficient to ensure 95% occupancy for any number of lines. The rate by which call arrivals exceed capacity is detected by setting triggers for Busy, No_Answer, and Route_Select_Failure events on calls admitted to the answering centre. (In practice triggers may also be set for Answer and Abandon to help with clean-up of call context information, though timeouts must always ensure old contexts are eventually removed). A monitor detects the rate of failure of calls admitted to the answering centre, and compares this rate with a target rate sufficient to ensure high occupancy (e.g. with 1.8 failures s.sup.-1). An adjustable rate control element determines the rate of calls admitted to the answering centre. If the rate of failures deviates from the target rate, a negative feedback mechanism adjusts the admission rate control element. The admission rate control sets the admitted call rate at a level which maintains the rate of failed calls close to the target rate for failures.

Detailed Description Text (37):
ACR monitors are dynamically-allocated resources which are allocated when a congestion event is encountered at an answering centre. Thus triggers must be set on at least a sample of calls sent to all answering centres if ACR is to be activated on overload. The sampling fraction may be small, e.g. 1 call in 10, as the purpose is to detect an overload to enable allocation of a monitor. Once a monitor has been allocated, a larger fraction of calls are sampled (e.g., 1 call in 2 or 1 call in 3) to enable sufficiently fast response from the control feedback loop.

Detailed Description Text (38):
ACR controls are dynamically-allocated resources which are allocated and given an initial leak rate when an ACR monitor first crosses an onset threshold. The leak rate of an allocated control is then adapted under negative feedback control from the monitor. When the overload abates the control continues to adapt to higher leak rates until it reaches a maximum value at which it is de-allocated.

Detailed Description Text (40):
In practice, ACR and ACS are found to be complementary in operation. When answering centres are receiving very high rates of ineffectives, then a significant reduction in the rate is achieved by ACR alone. When answering centres receive, for example, less than 1 ineffective every 10 seconds then ACR does not offer any reduction, while ACS offers a relatively small reduction. There is however a range which lies between these extremes where ACS offers a large reduction compared with the use of ACR on its own. Optionally, the OCS may monitor rates of ineffectives and invoke an ACS only when the rate lies in this optimum range.

Detailed Description Text (50):
The appendix below lists C source code for a prototype implementation of the ACS. The source code is part of a program which performs an event-based simulation of the operation of a number translation service. The service is implemented using an Intelligent Network. The C source code included here contains those components of the simulation program which are responsible for simulating operation of the service; of ACR based control of traffic to destinations; and of DACS (Dynamic ACS) based control of traffic to destinations. The code shown here also models the destinations themselves. Program components external to this fragment are responsible for managing the time-ordered event list of the event driven simulation; for modelling call arrivals; for reading the input data files containing parameters which control the simulation; and for printing and displaying simulation results.

Detailed Description Text (56):
The function answeringCentre( ) within this code models the destinations themselves in terms of the number of lines to the destination, the number of agents (people)

available to answer calls at the destination, call holding times, and the events in the progress of an individual call, as follows: calls admitted to a destination encounter busy if all lines are currently occupied. Otherwise, if an agent is available the call is answered immediately, and a CALL_COMPLETES event is created to model call disconnection. The CALL_COMPLETES event occurs after a call holding time which is negative-exponentially distributed with mean value given by a parameter of the service. If no agent is available the call may ring for a period set by RTNR_TIME. If an agent becomes available before the end of RTNR_TIME the call is answered, otherwise the call fails. When an event occurs during the progress of a call, the destination examines the flags component of the service data, to determine whether that event should be reported to the service. This action models the INAP message Request Report BCSM Event which would be sent to the source SSP in a real implementation. When one of the specified events occurs, the destination model creates a report of the state change which is loaded to the simulation's event list, for transmission to the service logic after a simulated delay. This delay simulates the signalling network and processing delays characteristic of a real IN. The function answeringCentre( ) is called when a specified event occurs. The event may be one of TS_COMPLETES, an attempt to initiate a new call; or CALL_COMPLETES, which occurs at the end of the conversation phase of a successful call; or RTNR_EVENT, which occurs when a call fails because no agent has become available to answer it within the specified time limit. The answeringCentre( ) maintains a count of callsBusy, i.e. those calls which are in an active conversation phase and occupying both a line and an agent; and a count of calls, i.e. those calls which are either in an active conversation phase or in a Ringing state. The difference, calls-callsBusy, is the number of calls which are in a Ringing state, and occupying a line but not an agent. These counts are manipulated appropriately whenever a call is initiated or a call changes its state due to an event.

Detailed Description Text (69):
The function service( ) simulates service logic. The tasks simulated are: call initiation, requests to OBCadmitCall( ) and ACSadmitCall( ) for permission to admit calls, and the processing of subsequent event reports for admitted calls. It is called with an event parameter: either TS_COMPLETES, AC_ANSWER, AC_RTNR, AC_COMPLETES, AC_BUSY, or AC_ABANDON, indicating respectively a new call, an event report for call answered, an event report for call not answered (ring tone no reply), an event report for call disconnection after a conversation phase, an event report for destination busy, and an event report for call abandoned (which is not implemented in this simple simulation). All of these calls to service( ) are made as a result of events on the simulation's time-ordered event list, as they occur after a delay following the occurrence of a previous event. TS_COMPLETES occurs after a delay simulating processing in preceding NIP functions. AC_ANSWER, AC-RTNR, AC_COMPLETES, AC_BUSY, and AC_ABANDON all occur as a result of events at the destination, but become known to the service only after a delay (simulating transmission across a signalling network).

CLAIMS:

4. A method according to claim 2, in which steps (a) to (d) are carried out only when the rate of calls to the destination number is above a first, lower threshold level and is below a second, upper threshold level.

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L9: Entry 11 of 22 File: USPT Nov 2, 1999

DOCUMENT-IDENTIFIER: US 5978465 A
TITLE: Method and apparatus for allocating resources in a call center

Application Filing Date (1):
19970505

Detailed Description Text (16):
If a communication link failure is identified at step 48, then the procedure returns to step 42 to copy the default resource allocation values to the current values. In this embodiment, the procedure copies all default values to the current values whenever a host communication link failure occurs. Thus, the call center will be reset to its default resource allocation, instead of continuing with its previous configuration (which may include one or more changes requested by the host). Since the communication link to the host has failed, the host will be unable to monitor or request changes in the call center resource allocation. Therefore, the default call center configuration is used rather than maintaining the previous version.

Detailed Description Text (34):
If the current inbound call volume does not exceed the second threshold in step 140 of FIG. 7, then the procedure continues to step 144 to continue processing inbound calls using existing inbound agents. The procedure then returns from step 142 or step 144 to step 132, thereby continually monitoring the current volume of inbound calls and transferring agents between inbound and outbound call processing, as needed. Thus, the procedure illustrated in FIG. 7 provides for the automatic control and allocation of call center resources based on the incoming call volume. Similar procedures can be used to automatically control and allocate other types of call center resources.

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L9: Entry 12 of 22 File: USPT Aug 17, 1999

DOCUMENT-IDENTIFIER: US 5940753 A
TITLE: Controller for cellular communications system

Application Filing Date (1):
19971006

Detailed Description Text (3):
Referring now to FIG. 1(a), an overview of a communications system 10 is presented showing the functional inter-relationships of the major elements. The system network control center 12 directs the top level allocation of calls to satellite and ground regional resources throughout the system. It also is used to coordinate system-wide operations, to keep track of user locations, to perform optimum allocation of system resources to each call, dispatch facility command codes, and monitor and supervise overall system health. The regional node control centers 14, one of which is shown, are connected to the system network control center 12 and direct the allocation of calls to ground nodes within a major metropolitan region. The regional node control center 14 provides access to and from fixed land communication lines, such as commercial telephone systems known as the public switched telephone network (PSTN). The ground nodes 16 under direction of the respective regional node control center 14 receive calls over the fixed land line network, encode them, spread them according to the unique spreading code assigned to each designated user, combine them into a composite signal, modulate that composite signal onto the transmission carrier, and broadcast them over the cellular region covered.

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L9: Entry 19 of 22 File: JPAB Aug 21, 1998

DOCUMENT-IDENTIFIER: JP 10224478 A
TITLE: DEVICE AND METHOD FOR AUTOMATICALLY ALLOCATING CALL CENTER AGENT TO SKILL IN CALL CENTER

Abstract Text (2):
SOLUTION: An agent vector monitors the selected performance parameter of call center such as service time, time in queue, call volume, call abandon rate, profit provided by processing calls requiring different skills through different agents and the ratio of work to be consumed by agents for processing the calls requiring different skills (402 and 404) and in order to optimize a prescribed target (406), by changing the skill to which the agent is allocated (logged-in) or changing the relative priority (speciality level) of the skill of agent, for example, the call processing allocation of agent is automatically adjusted.

Application Date (1):
19980127

WEST

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L9: Entry 21 of 22 File: EPAB Nov 25, 1999

DOCUMENT-IDENTIFIER: WO 9960766 A1
TITLE: (<u>HUMAN</u>) RESOURCE ALLOCATION IN (<u>CALL CENTRE</u>) TASK MANAGEMENT

Abstract Text (1):
A (call centre) telephone call (handling) management system (64) is configured to optimise allocation of appropriately skilled agent operators (50), for call handling, in providing a predetermined "level of service"; by monitoring (36) call density and deploying knowledge (45, 46) of agent resource skill profiles, availability and attendant budget limitations and applying those factors to a modelling algorithm - allowing call allocation across agent skill groups, or even on a one-to-one basis with individual agents.

Application Date (1):
19990518